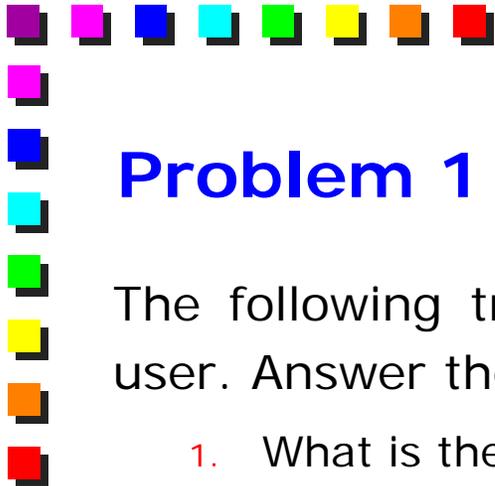


Problems on Voice over IP (VoIP)



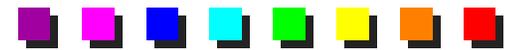


Problem 1

The following trace refers to the registration phase of a SIP user. Answer the following questions:

1. What is the IP address of the SIP client?
2. What is the IP address of the SIP proxy?
3. Why the first registration attempt does fail?
4. provide a brief explanation of the whole registration process

N.	Network	Transport	Application
1	IP: 130.192.225.36 => 130.192.225.79 (Len 60)	UDP: port 4136 => 53	DNS Query
2	IP: 130.192.225.79 => 130.192.225.36 (Len 328)	UDP: port 53 => 4136	DNS Response
3	IP: 130.192.225.36 => 130.192.225.79 (Len 601)	UDP: port 63772 => 5060	SIP: REGISTER sip:ipv6.polito.it SIP/2.0
4	IP: 130.192.225.79 => 130.192.225.36 (Len 728)	UDP: port 5060 => 63772	SIP: SIP/2.0 401 Unauthorized
5	IP: 130.192.225.36 => 130.192.225.79 (Len 800)	UDP: port 63772 => 5060	SIP: REGISTER sip:ipv6.polito.it SIP/2.0
6	IP: 130.192.225.79 => 130.192.225.36 (Len 706)	UDP: port 5060 => 63772	SIP: SIP/2.0 200 OK



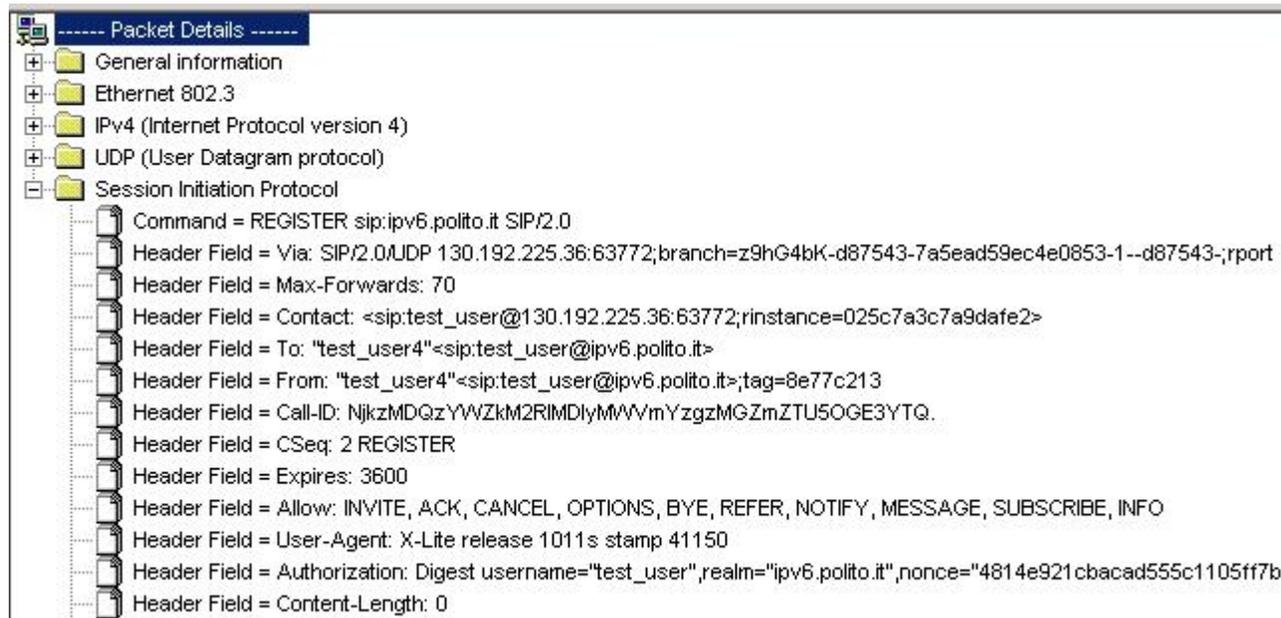
Problem 1 - answers

1. The IP address of the UA SIP is 130.192.225.36
2. The IP address of the SIP proxy is 130.192.225.79
3. The first registration attempt fails because SIP UA did not include credentials for authentication in the REGISTER message
4. The registration procedure is required for
 1. Authenticating a user that tries to access a SIP domain
 2. Associating the SIP URI with the SIP UA (host) where the user is connected
 1. In this way, the user can be reached using his SIP URI
 2. If the user moves to a different IP address, he should re-register with his domain SIP server

N.	Network	Transport	Application
1	IP: 130.192.225.36 => 130.192.225.79 (Len 60)	UDP: port 4136 => 53	DNS Query
2	IP: 130.192.225.79 => 130.192.225.36 (Len 328)	UDP: port 53 => 4136	DNS Response
3	IP: 130.192.225.36 => 130.192.225.79 (Len 601)	UDP: port 63772 => 5060	SIP: REGISTER sip:ipv6.polito.it SIP/2.0
4	IP: 130.192.225.79 => 130.192.225.36 (Len 728)	UDP: port 5060 => 63772	SIP: SIP/2.0 401 Unauthorized
5	IP: 130.192.225.36 => 130.192.225.79 (Len 800)	UDP: port 63772 => 5060	SIP: REGISTER sip:ipv6.polito.it SIP/2.0
6	IP: 130.192.225.79 => 130.192.225.36 (Len 706)	UDP: port 5060 => 63772	SIP: SIP/2.0 200 OK

Problem 2

Explain the meaning and the role of the main SIP headers of the following message



The screenshot displays the 'Packet Details' window of a network analyzer, showing the structure of a SIP REGISTER message. The tree view on the left includes 'General information', 'Ethernet 802.3', 'IPv4 (Internet Protocol version 4)', 'UDP (User Datagram protocol)', and 'Session Initiation Protocol'. The main pane lists the following fields:

- Command = REGISTER sip:ipv6.polito.it SIP/2.0
- Header Field = Via: SIP/2.0/UDP 130.192.225.36:63772;branch=z9hG4bK-d87543-7a5ead59ec4e0853-1--d87543-;rport
- Header Field = Max-Forwards: 70
- Header Field = Contact: <sip:test_user@130.192.225.36:63772;rinstance=025c7a3c7a9dafe2>
- Header Field = To: "test_user4"<sip:test_user@ipv6.polito.it>
- Header Field = From: "test_user4"<sip:test_user@ipv6.polito.it>;tag=8e77c213
- Header Field = Call-ID: NjkzMDQzYWZkM2RlMDlyMmVhYzgzMGZmZTU5OGY3YTQ.
- Header Field = CSeq: 2 REGISTER
- Header Field = Expires: 3600
- Header Field = Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
- Header Field = User-Agent: X-Lite release 1011s stamp 41150
- Header Field = Authorization: Digest username="test_user",realm="ipv6.polito.it",nonce="4814e921cbacad555c1105ff7b
- Header Field = Content-Length: 0

Problem 2 - solution

----- Packet Details -----

- General information
- Ethernet 802.3
- IPv4 (Internet Protocol version 4)
- UDP (User Datagram protocol)
- Session Initiation Protocol
 - Command = REGISTER sip:ipv6.polito.it SIP/2.0
 - Header Field = Via: SIP/2.0/UDP 130.192.225.36:63772;branch=z9hG4bK-d87543-7a5ead59ec4e0853-1--d87543-;rport
 - Header Field = Max-Forwards: 70
 - Header Field = Contact: <sip:test_user@130.192.225.36:63772;instance=025c7a3c7a9daf2>
 - Header Field = To: "test_user4"< sip:test_user@ipv6.polito.it>
 - Header Field = From: "test_user4"< sip:test_user@ipv6.polito.it>;tag=8e77c213
 - Header Field = Call-ID: NjkzMDQzYWVZkm2RlMDlyMwVvMmYzgzMGZmZTU5OGE3YTQ.
 - Header Field = CSeq: 2 REGISTER
 - Header Field = Expires: 3600
 - Header Field = Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
 - Header Field = User-Agent: X-Lite release 1011s stamp 41150
 - Header Field = Authorization: Digest username="test_user",realm="ipv6.polito.it",nonce="4814e921cbacad555c1105ff7b"
 - Header Field = Content-Length: 0

The "To" header stores the URI identifying the SIP user

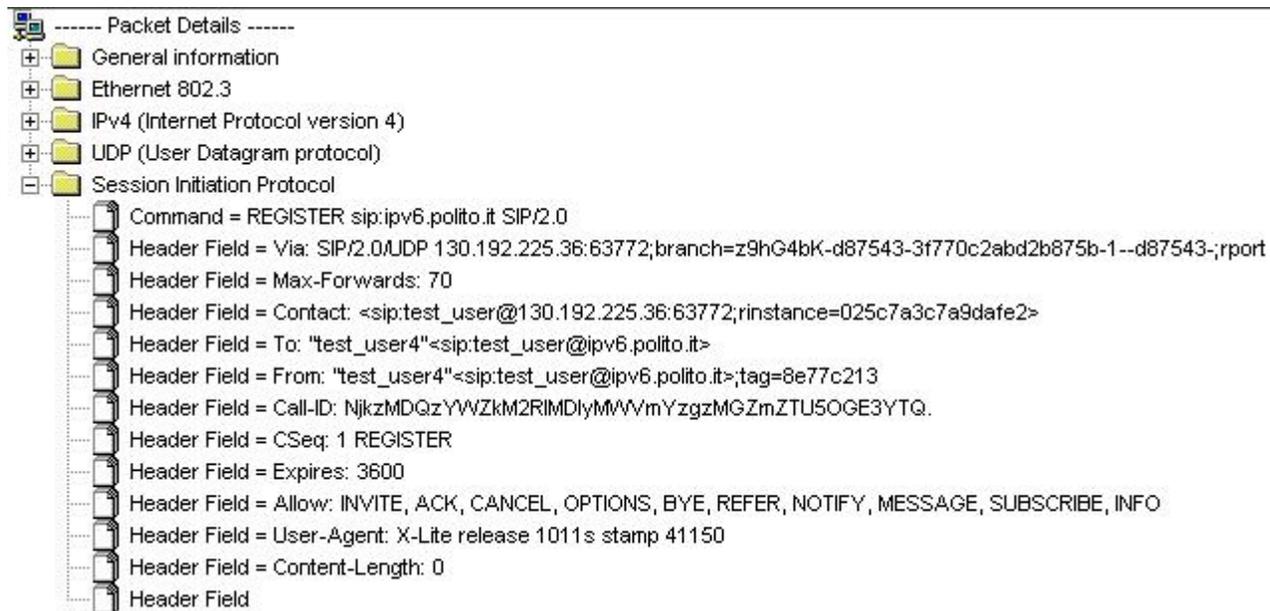
Period of validity of the registration

Authentication credentials

The "contact" header stores the information about the current position of the user (IP+port number)

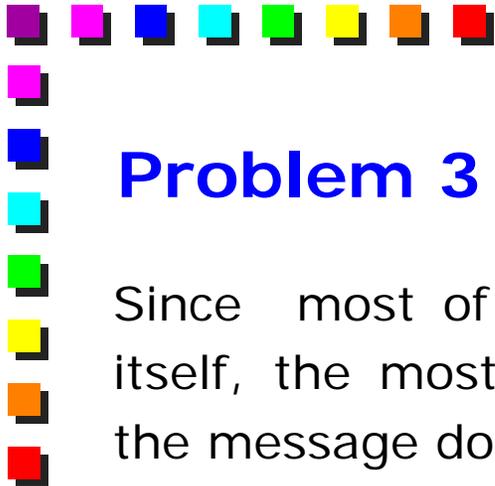
Problem 3

Given the following SIP REGISTER message, assuming a correct configuration for the SIP client, what could be the answer to this request?



The screenshot shows the 'Packet Details' window for a SIP REGISTER message. The tree view on the left is expanded to show the 'Session Initiation Protocol' section. The details are as follows:

- Command = REGISTER sip:ipv6.polito.it SIP/2.0
- Header Field = Via: SIP/2.0/UDP 130.192.225.36:63772;branch=z9hG4bK-d87543-3f770c2abd2b875b-1--d87543-;rport
- Header Field = Max-Forwards: 70
- Header Field = Contact: <sip:test_user@130.192.225.36:63772;rinstance=025c7a3c7a9dafa2>
- Header Field = To: "test_user4"<sip:test_user@ipv6.polito.it>
- Header Field = From: "test_user4"<sip:test_user@ipv6.polito.it>;tag=8e77c213
- Header Field = Call-ID: NjkzMDQzYWZkM2RlMDlyMmVhYzgzMGZmZTU5OGE3YTQ.
- Header Field = CSeq: 1 REGISTER
- Header Field = Expires: 3600
- Header Field = Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
- Header Field = User-Agent: X-Lite release 1011s stamp 41150
- Header Field = Content-Length: 0
- Header Field



Problem 3 - answer

Since most of the SIP server require a user to authenticate itself, the most likely answer is "401 Unauthorized", because the message does not include credentials for the authentication.

No username – IP association will be created and the server response will include a challenge for the authentication.



Problem 4

The following is a trace of an INVITE session, assuming that the caller is in the SIP domain "ipv6.polito.it", answer the following questions:

1. What is the username of both clients?
2. What is the IP address and port number for both clients?
3. What is the meaning of the "100 Trying" message?
4. What is the meaning of the "180 Ringing" message?
5. Record-routing is enabled in the SIP proxy(ies)?
6. What is the minimum number of Sip proxies that can be traversed by an INVITE message?

N.	Network	Application
1	IP: 130.192.225.135 => 130.192.225.79 (Len 60)	DNS Query
2	IP: 130.192.225.79 => 130.192.225.135 (Len 328)	DNS Response
3	IP: 130.192.225.135 => 130.192.225.79 (Len 768)	SIP: INVITE sip:test_user@ipv6.polito.it SIP/2.0
4	IP: 130.192.225.79 => 130.192.225.135 (Len 604)	SIP: SIP/2.0 100 trying -- your call is important to us
5	IP: 130.192.225.79 => 130.192.225.135 (Len 481)	SIP: SIP/2.0 180 Ringing
6	IP: 130.192.225.79 => 130.192.225.135 (Len 811)	SIP: SIP/2.0 200 OK
7	IP: 130.192.225.135 => 130.192.225.79 (Len 509)	SIP: ACK sip:test_user@130.192.225.36:63772;rinstance=025c7a3c7a9da...
8	IP: 130.192.225.79 => 130.192.225.135 (Len 655)	SIP: BYE sip:livio@130.192.225.135:7226 SIP/2.0
9	IP: 130.192.225.135 => 130.192.225.79 (Len 495)	SIP: SIP/2.0 200 OK

Problem 4: answers (1/3)

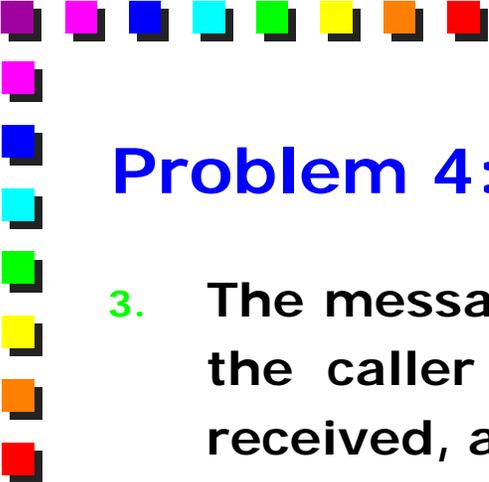
N.	Network	Application
1	IP: 130.192.225.135 => 130.192.225.79 (Len 60)	DNS Query
2	IP: 130.192.225.79 => 130.192.225.135 (Len 328)	DNS Response
3	IP: 130.192.225.135 => 130.192.225.79 (Len 768)	SIP: INVITE sip:test_user@ip6.polito.it SIP/2.0
4	IP: 130.192.225.79 => 130.192.225.135 (Len 604)	SIP: SIP/2.0 100 trying -- your call is important to us
5	IP: 130.192.225.79 => 130.192.225.135 (Len 481)	SIP: SIP/2.0 180 Ringing
6	IP: 130.192.225.79 => 130.192.225.135 (Len 811)	SIP: SIP/2.0 200 OK
7	IP: 130.192.225.135 => 130.192.225.79 (Len 509)	SIP: ACK sip:test_user@130.192.225.36:63772 instance=025c7a307a9da...
8	IP: 130.192.225.79 => 130.192.225.135 (Len 655)	SIP: BYE sip:livio@130.192.225.135:7226 SIP/2.0
9	IP: 130.192.225.135 => 130.192.225.79 (Len 495)	SIP: SIP/2.0 200 OK

"livio"

2. Caller address: 130.192.225.135:7226

Called address: 130.192.225.36:63772

- IP addresses and port numbers can be read from the messages ACK and BYE (respectively)
- Once the dialog is established, the two UAs can communicate directly, using the addresses and port numbers discovered during the INVITE process



Problem 4: answers (2/3)

3. The message "100 Trying" is sent from the proxy to the caller UA to signal that the request has been received, and it is under process
 4. The message "180 Ringing" is sent from the called UA to the caller one (through the proxies) to confirm the reception of the request and for notifying the caller that the called phone is now ringing
 - Only when the human user will take the phone "off-hook" the response **200 OK** is generated
- 

Problem 4: answers (3/3)

N.	Network	Application
1	IP: 130.192.225.135 => 130.192.225.79 (Len 60)	DNS Query
2	IP: 130.192.225.79 => 130.192.225.135 (Len 328)	DNS Response
3	IP: 130.192.225.135 => 130.192.225.79 (Len 768)	SIP: INVITE sip:test_user@ipv6.polito.it SIP/2.0
4	IP: 130.192.225.79 => 130.192.225.135 (Len 604)	SIP: SIP/2.0 100 trying -- your call is important to us
5	IP: 130.192.225.79 => 130.192.225.135 (Len 481)	SIP: SIP/2.0 180 Ringing
6	IP: 130.192.225.79 => 130.192.225.135 (Len 811)	SIP: SIP/2.0 200 OK
7	IP: 130.192.225.135 => 130.192.225.79 (Len 500)	SIP: ACK sip:test_user@130.192.225.36:63772;rinstance=025c7a3c7a9da...
8	IP: 130.192.225.79 => 130.192.225.135 (Len 655)	SIP: BYE sip:livio@130.192.225.135:7226 SIP/2.0
9	IP: 130.192.225.135 => 130.192.225.79 (Len 495)	SIP: SIP/2.0 200 OK

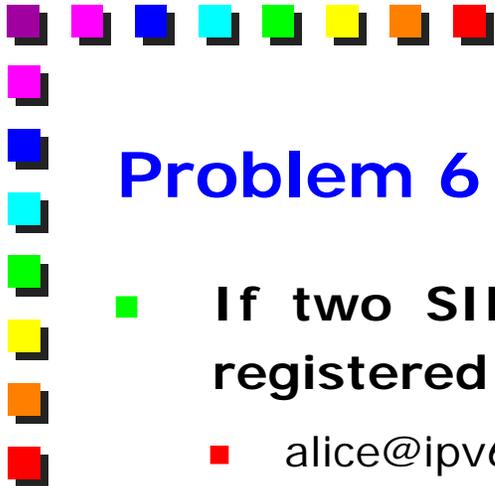
5. It is possible to see that the proxy is always involved in all the SIP messages. Hence record routing is enabled.
6. Since both users belong to the same SIP domain, the minimum number of proxies traversed is 1 (it would be 2 if they were in 2 different domains)

Problem 5 (without solution)

The following is a trace of an INVITE session, assuming that the caller is in the SIP domain "ipv6.polito.it", answer the following questions:

- What is the IP address and port number of both SIP UA?
- Is record routing enabled?

No. -	Time	Source	Destination	Protocol	Info
1	0.000000	130.192.225.79	130.192.225.135	SIP/SD	Request: INVITE sip:livio@130.192.225.135:7225
2	0.021484	130.192.225.135	130.192.225.79	SIP	Status: 180 Ringing
3	2.746769	130.192.225.135	130.192.225.79	SIP/SD	Status: 200 OK, with session description
4	2.855290	130.192.225.36	130.192.225.135	SIP	Request: ACK sip:livio@130.192.225.135:7225
5	8.651164	130.192.225.135	130.192.225.36	SIP	Request: BYE sip:test_user@130.192.225.36:7884
6	8.754173	130.192.225.36	130.192.225.135	SIP	Status: 200 OK



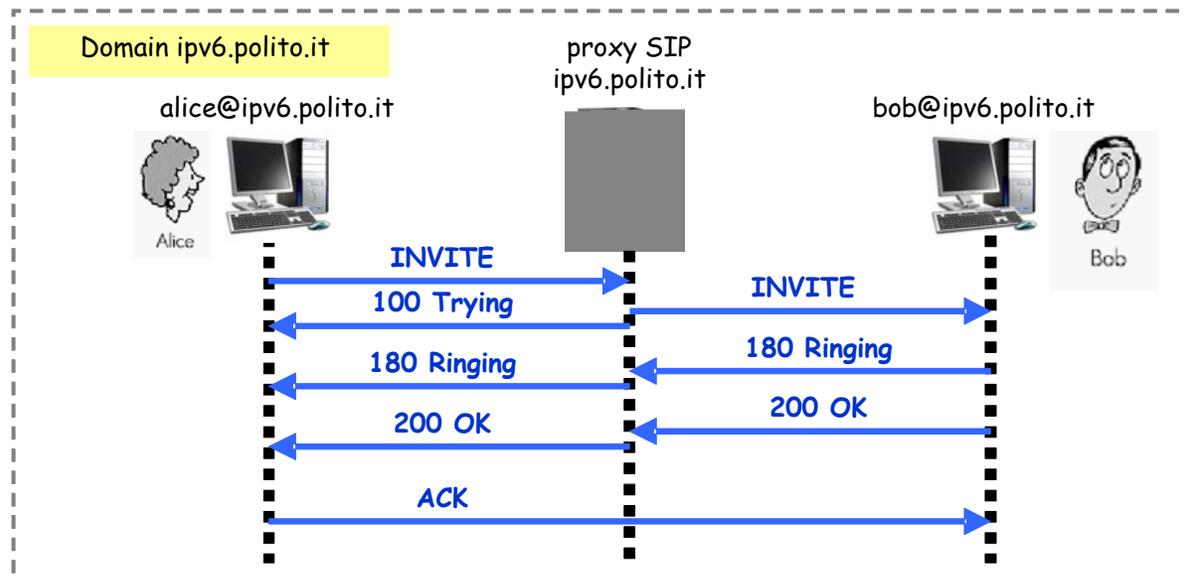
Problem 6

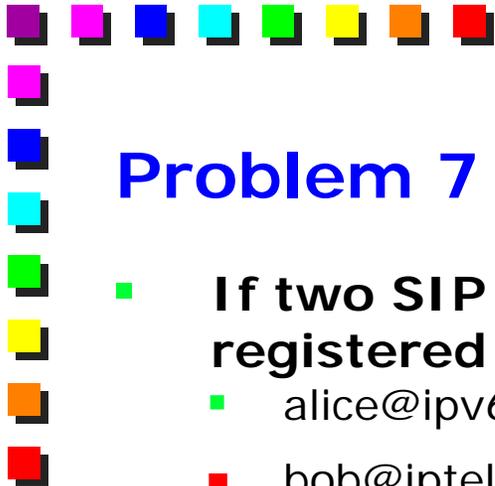
- If two SIP users have the following URI correctly registered in their domain:
 - `alice@ipv6.polito.it`
 - `bob@ipv6.polito.it`
- 1. Draw a diagram with all the messages exchanged between Alice and Bob, in order to set up a call, including:
 1. Possible auxiliary messages
 2. Messages sent by the proxy



Problem 6 - answer

- The requested diagram is shown below
 - since the registration has already taken place, both UAs can access the proxy without additional DNS queries, but taking advantage of their DNS caches





Problem 7

- **If two SIP users have the following URI correctly registered in their domain**
 - `alice@ipv6.polito.it`
 - `bob@iptel.org`
- 1. **Draw a diagram with all the messages exchanged between Alice and Bob, in order to set up a call, including:**
 1. Possible auxiliary messages
 2. Messages sent by the proxies



Esercizio 7 - soluzione

